EC 2302 - Digital Signal Processing

Two Mark Questions

UNIT - I :-

1. What are the applications of DSP?

* Speech and musical processing
* Navigation and secured communication in military.
* Noise removal, echo cancellation and processing of satellite signal.
* Spectrum analysis, Robot and process control.
* Image processing
* X-ray, CT and MRI Image analysis, EEG signal processing.

2. What are the advantages of DSP?

* Greater accuracy
* High speed
* Flexibility in configuration
* Ease of storage of data.
* Implementation of sophisticated algorithm.

3. What are the ~~dind~~ disadvantages of DSP?

* Quantisation errors
* System complexity
* Bandwidth is limited by sampling rate.
* Transmission power is high.

4. Define DFT.

$$X(k) = \sum_{n=0}^{N-1} x(n) \cdot e^{-j2\pi kn/N}$$

Inverse DFT is

$$x(n) = \frac{1}{N} \sum_{k=0}^{N-1} X(k) \cdot e^{j2\pi kn/N}$$

5. Compute the DFT of i) $x(n) = \delta(n)$ (ii) $x(n) = \delta(n-n_0)$

i) $X(k) = 1$ (ii) $X(k) = e^{-j2\pi n_0 k/N}$

6. Differentiate DFT and FFT.

Refer class notes.

7. What is the radix-2 FFT?

The number of stages is expressed in terms of the integer power of 2.

$$\boxed{N = 2^L}$$

where $L \to$ no. of stages.

8. Draw basic DIT butterfly diagram.

Refer class notes.

9. How many multiplications and additions are involved in radix-2 FFT?

No. of multiplications $= N/2 \log_2 N$

No. of additions $= N \log_2 N$

10. What is phase factor or twiddle factor?

$$W_N = e^{-j2\pi/N}$$

11. Distinguish between DFT and DTFT.

DFT	DTFT
i) Discrete frequency spectrum	i) continuous frequency spectrum.
ii) Obtained by performing sampling operation in both time and frequency domains.	ii) Sampling is performed only in time domain.

12. What is meant by 'in place of' computation?

In N point FFT algorithm, the results of each stage are stored in the same 2N registers. This is called 'in place of' computation.

13. Compute the value of twiddle factor W_{64}^{16}.

$$W_N = e^{-j2\pi/N}$$

$$W_{64} = e^{-j2\pi/64}$$

$$W_{64}^{16} = e^{-j32\pi/64} = e^{-j\pi/2} = -j$$

14. What is zero padding ? What is its use?

To find the N-point DFT of the sequence $x(n)$ of length 'L', $(N-L)$ zeros are appended to $x(n)$. This is known as zero padding.

Uses :-

i) Better display of the frequency spectrum

ii) DFT can be used in linear filtering.

15. Define circular convolution.

The circular convolution between $x_1(n)$ and $x_2(n)$ is

$$x_3(m) = \sum_{n=0}^{N-1} x_1(n) \cdot x_2((m-n))_N$$

16. State the differences between overlap add method and overlap save method.

Refer class notes.

UNIT - II

1. Differences between Butterworth and Chebyschev filters.

Refer class notes.

2. Properties of Butterworth filter. } from the Q.No. 1.

3. Properties of Chebyschev filter. }

4. Give the steps in the design of a digital filter from analog filters.

i) Map the desired digital filter specifications into those for an equivalent analog filter.

ii) Derive the analog transfer function for the analog prototype.

iii) Transform the transfer function of the analog prototype into an equivalent digital filter transfer function.

5. What is impulse invariant mapping? What are its limitations?

In this method, the impulse response of resulting digital filter is a sampled version of the impulse response of the analog filter.

The transfer function of the digital filter can be obtained using the transformation

$$\frac{1}{s-p_k} \longrightarrow \frac{1}{1-e^{p_k T} z^{-1}}$$

$$\text{ie) } H(z) = \sum_{k=1}^{N} \frac{c_k}{1-e^{p_k T} z^{-1}}$$

Limitation :- In Impulse invariant mapping an infinite no. of poles in s-plane are mapped into same location in z-plane and thus the aliasing effect is produced.

6. Give the frequency transform relation for converting low pass into band pass in analog domain.

Refer class notes - Frequency Transformation in analog domain.

7. Compare Impulse Invariant over Bilinear transformation

Refer class notes.

8. What is warping effect? Refer class notes.

9. What is prewarping? Refer class notes.

10. What are the disadvantages of direct form?

The direct form realization is extremely sensitive to parameter quantization. When the order of the system N is large, a small change in a filter coefficient due to parameter quantization results in a large change in the location of the poles and zeros of the system.

11. What is the advantage of direct form II?

The no. of memory locations required is less than that of direct form I.

12. Advantage of cascade realization.

Quantization errors can be minimized if a system is realized in cascade form.

UNIT-III :-

1. State the properties of FIR filter.
2. Differentiate FIR and IIR.

} Refer class notes.

3. Advantages and disadvantages of FIR filter.

Refer class notes.

4. What is meant by linear phase characteristic? What is the necessary and sufficient condition for linear phase charac. of an FIR filter.

For a linear phase filter $\theta(\omega) \propto \omega$.

The linear phase filter does not alter the shape of the original signal.

$$h(n) = h(N-1-n)$$

Group delay = const.

Phase delay = const.

5. Define Hamming and Hanning window.

Refer class notes.

6. What are the uses of symmetric impulse response and antisymmetric impulse response.

Symmetric impulse response with M, odd } — To design LPF, HPF, BPF and BRF

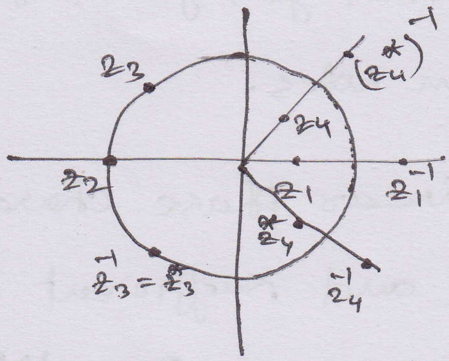
Symm. imp. response with M even — To design LPF and BPF

Antisymm. imp. resp. — To design Hilber transform and differentiators

7. Why FIR is always stable?

All poles are at the origin.

8. How zeros in FIR filter are located?



iv) z_4 is a complex zero. This group contains four zeros $z_4, z_4^{-1}, z_4^*, (z_4^*)^{-1}$.

i) z_1 is a real zero with $|z_1| < 1$, then z_1^{-1} is also a real zero and there are two zeros in this group.

ii) $z_2 = -1$, then $z_2^{-1} = z_2$ and this group contains only one ze

iii) z_3 is a complex zero, then $z_3^{-1} = z_3^*$ and two zeros in this group.

9. What are the disadvantages of Fourier Series Method?

10. What is Gibb's phenomenon? or

What are Gibb's oscillation?

Refer class notes.

11. What is window? Why it is necessary?

Abrupt truncation of the Fourier Series will lead to oscillations in the pass band and stop band. These oscillations can be reduced through the use of less abrupt truncation of the Fourier Series. This can be achieved by multiplying the infinite response by a finite weighing sequence ~~w~~ $w(n)$, called a window.

12. What are the desirable characteristics of window?

* Central lobe of the freq. response of the window should contain most of the energy and should be narrow.
* The highest side lobe level of the freq. response should be small.
* The side lobes of the frequency response should decrease in energy rapidly as ω tends to π.

13. For what type of filters freq. sampling method is suitable?

For narrow band frequency selective filters.

UNIT- IV :-

1. What is meant by finite word length effects? List some of the finite word length effects in digital filters.

Refer class notes.

2. What are the different formats of fixed point representation?

i) Sign magnitude
ii) One's complement
iii) Two's complement

3. What are the types of arithmetic used in dig. computer?

* Floating point arithmetic
* 2's complement arithmetic.

4. Compare fixed point and floating point representations.

Refer class notes.

5. What are the types of quantization employed in digital computer?

* Truncation
* Rounding.

6. What are truncation and rounding?

Refer class notes.

7. What is the range of rounding error? Refer class notes.

8. What are limit cycles? What are its types?

Refer class notes.

9. What are zero input limit cycle and overflow limit cycle? Refer class notes.

10. What is dead band? Refer class notes.

11. How overflow limit cycle can be eliminated?

By using saturation arithmetic or by scaling input signal to the adder.

12. What is saturation arithmetic?

When the result of an arithmetic operation exceeds the dynamic range of number system, then the result is set to maximum or minimum possible value.

13. What is the drawback in saturation arithmetic?

It introduces non-linearity in the adder which creates signal distortion.

14. Definition of quantization error, coefficient quantization error, product quantization error, overflow error.

Refer class notes.

15. Represent the given Nos. in floating point format.

a) $+0.25_{10} = 0.010$

$= 0.100 \times 2^{-1}$

$= 0.100000 \times 2^{101}$

b) $-0.25 = 1.010$

$= 0.101 \times 2^{1}$

$= 0.101000 \times 2^{001}$

UNIT - V :-

1. What is multirate DSP?
2. What are decimation and interpolation?

} Refer class notes.

3. Why interpolation is followed by decimation?

* The filters of interpolation and decimation have the same cut off frequency. So, a single low pass filter can be used instead of two filters.

* To preserve the desired spectral characteristics of the input signal.

4. Give any 4 application of multirate DSP.

* Sub-band coding
* Image compression
* High quality digital audio and video systems.
* Digital storage systems.

5. Write some advantages of multirate processing.

* The reduction in no. of computations
* The reduction in memory requirement
* The reduction in finite word length effects.

6. Decimator – The device which performs the process of decimation.

Symbolic representation → $x(n) \rightarrow \boxed{\downarrow D} \rightarrow y(n) = x(Dn)$

7. What is anti-aliasing filter?

The low pass filter used at the input of decimator is called anti-aliasing filter. It is used to limit the bandwidth of an input signal to π/D in order to prevent the aliasing of output spectrum of decimator ~~of~~ for decimation by D.

8. Interpolator – The device which performs the process of interpolation

<u>Symbolic representation</u>

$$x(n) \longrightarrow \boxed{\uparrow I} \longrightarrow y(n) = x(n/I)$$

9. <u>Anti-Imaging filter:-</u>

The low pass filter used at the output of an interpolator is called anti-imaging filter. It is used to eliminate the multiple images in the output spectrum of the interpolator.

10. What is polyphase decomposition?

The process of dividing a filter into a no. of sub-filters which differ only in phase characteristics is called polyphase decomposition.

11. Quadrature Mirror Filter – A two channel subband coding